MIX2000

4 inputs portable mixer

User Manual

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1. Introduction

The MIX2000 is a portable mixer designed especially for outdoor recording (ENG). It is adequate for cinema production as well, thanks to its audio performance and its M/S format handling capabilities.

The mixer features four symmetrical microphone/line inputs with microphone powering, and a stereo output with limiter/compressor.

As noticeable highlights, the mixer features:

- Low weight and size (2 kg, 240 x 190 x 43 mm)
- Ultra-low noise (-126 dBu)
- Outstanding input headroom (56 dB)
- Fully M/S compatible
- VCA controlled master stage with selectable compressor and limiter
- 4 symmetrical inputs with selectable high-pass filters
- Wide scale LED bar-graphs with adjustable brightness
- High performance headphone amplifier with source and listening mode selection

2. Functions

The general synoptic diagram shows the functions of the mixer. It can be found in annex 6.2, General synoptic diagram (page 32).

2.1. Input stages

Each input is symmetrical, transformer coupled.

Microphone powering can be inserted, in phantom mode or "ton-adder" mode[1] (12V). In phantom mode, the power supply is 12V or 48V (internally configurable).

Line level is supported thanks to a –26 dB pad. For a microphone input, the input gain is adjustable by 6 dB steps over a 42 dB range.

2.2. Input channel processing

Each channel can be applied a selectable 12 dB/octave high-pass filter (Off / 80 Hz / 120 Hz cut-off frequency), and phase inversion.

An LED indicates possible channel overload.

The channel feeds the two mixer buses through a channel fader and a panoramic potentiometer.

Each channel pair (1-2 or 3-4) can be coupled so that the even-numbered channel fader simultaneously controls the level for both channels.

2.3. Main channels (MASTER)

The mixer buses feed the main outputs through VCA-based fader and balance adjustments.

[1] Not available on some equipment versions

A limiter or compressor can be inserted in the Master channel. This limiter/compressor features two selectable operating modes:

- Normal stereo mode, where the limiter/compressor applies same gain to both mixing buses;
- “Independent” mode, where each channel is processed separately, to be used when the main outputs are used as two mono outputs. This mode can be selected by changing the internal configuration (for such change, refer to chapter 4.2).

2.4. Outputs (MAIN and SUB)

The Master channels are available in symmetrical format on the MAIN outputs, with a –60 dB pad separately selectable on each output.

The mixer buses are available in asymmetrical format on the SUB outputs, with a selectable –60 dB pad. The output gain is adjustable in 6 dB steps thanks to a 6 position switch (0 to –30 dB attenuation from normal gain).

It is possible to encode the MAIN outputs in M/S format (or decode to L/R stereo if the mixer is used for recording with M/S stereo microphones). In such case, matrixing is applied as follows:
“Left” MAIN output = MAIN L + MAIN R *(where MAIN L/R are the signals before matrixing)*
“Right” MAIN output = MAIN L - MAIN R

2.5. Monitoring

A signal in the mixer can be selected for stereo monitoring on a headphone and a programme level meter.

The monitored signal can be selected among the following sources:

- Inputs 1 and 2;
- Inputs 3 and 4;
- SUB outputs;
- MAIN outputs;
- Return signal (from the “Monitor” interface).

In addition, a Solo push-button on each input channel enables direct pre-fader listening of this input. When pressing the Solo button, the signal is monitored in mono mode (signal sent to both ears and both meter channels).

The level meter is a wide scale (-40 to +8 VU) two-channel bar-graph display, on 16 LED per channel. Depending on the internal configuration of the mixer, the level detection has VU or PPM ballistics. In both cases, two level display modes are available:

- In the standard mode, the level is displayed in bar-graph (“thermometer”) mode, and the last peak is held for a while on a separate dot (“peak-hold” function);
- In the “dot-only” mode, only the last peak is displayed (peak-hold). This mode can be used e.g. for getting a more discreet display.

The level display also has adjustable brightness.

Various listening modes are available on the headphone output:

- Normal stereo or reverse stereo;
- Right signal only or left signal only on both ears;
- Mono (left/right summing) mode, for checking mono compatibility, phase coherence, etc.

Last, M/S↔L/R matrixing can be used for monitoring. This is used for listening to a normal stereo signal in the headphone while using M/S microphones[1]. As an alternate use, when the mixer is operated in normal L/R stereo mode, M/S encoding of the monitored signal can be used for checking the stereo correlation (checking the "S" signal level in relation with the "M" signal level).

Whenever used, matrixing is applied as follows:

"Left" matrix output = SIG L + SIG R (SIG L/R are the monitored signals before matrixing)
"Right" matrix output = SIG L - SIG R

2.6. Internal oscillator

An internal stereo oscillator is available for sound checking and alignment. The oscillator frequency is 1000 Hz on the left and 400 Hz (intermittent) on the right. By internal configuration, it is possible to have 1 kHz both on left and right channels. It is also possible to make the right signal permanent.

2.7. Slate

A talkback microphone, integrated in the mixer, can be used for inserting the operator's voice on a mixer output. Depending on the internal configuration of the mixer, the slate signal is inserted on one (or more) of the following outputs:

- Left MAIN output;
- Left SUB output;
- Output to RF transmitter;
- Left MON output.

2.8. Monitor (MON)

The mixer features a MONITOR interface, typically used for intercommunication with a camcorder or a control room (outside broadcasting applications, ENG). This interface includes the following signals:

- Stereo "MON OUT" output (MAIN signal);
- Stereo "MON Return" signal (input to the mixer for monitoring);
- Slate signal (output);
- "On-air" control input.

[1] If the MAIN signal has been applied M/S↔L/R matrixing at the output, the MAIN signal is automatically monitored with M/S decoding, because the signal is picked up after the M/S↔L/R matrix.

2.9. Power supply

The mixer operates from disposable or rechargeable batteries (8 AA-size batteries). It can be operated from an external DC source as well.

The mixer can automatically switchover from one power source to the other, with no interruption or noise.

2.10. Miscellaneous

The mixer includes a dedicated output for feeding a wireless RF transmission device, including power supply for the RF transmitter.

Lastly, extension interfaces are available for coupling two (or more) MIX2000 mixers together, for e.g. increasing the number of inputs.

2.11. Digital output (option)

The MIX2000 can be optionally[1] equipped with an analogue to digital converter (24 bit resolution). This module converts the MAIN output signals and outputs a stereo digital signal, selectable in AES/EBU format or SPDIF (IEC958).

The analogue MAIN outputs are still available when the "digital output" option is installed.

[1] This option is not displayed on the synoptic diagram

3. Operating mode – Detailed description

3.1. Left panel: "Inputs"

This panel includes the input connectors and input settings. The "Monitor" interface is also available on this side, including the interface connector and a potentiometer for adjusting the Monitor Return gain.

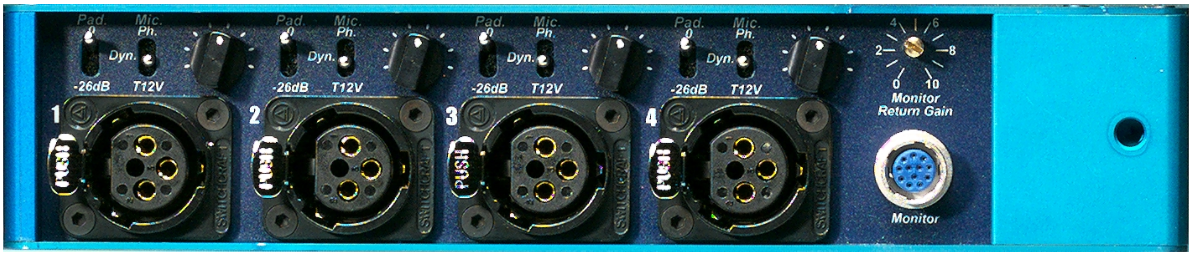


Figure 1 - Left panel – Inputs and Monitor interface

The following elements are available for each input:

- Input connector: female XLR;
- Microphone power selection switch: **Dyn** (dynamic microphone or line level); **Ph.** (phantom power supply), **T12V**[1] ("Tonadder");
- Pad switch;
- Input gain selection switch, with 8 positions.

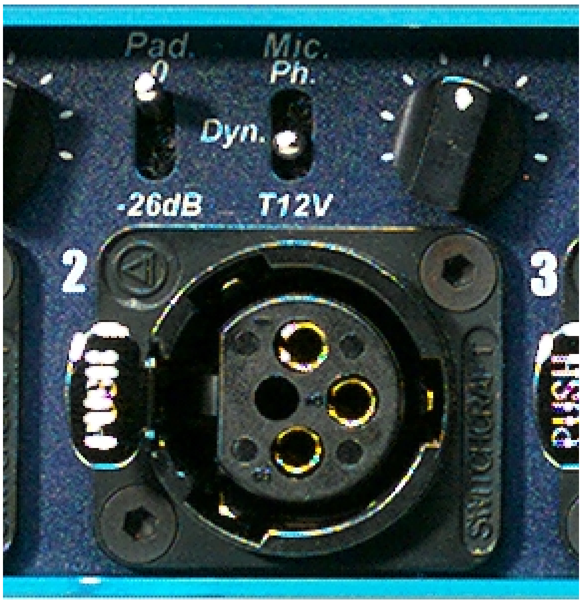


Figure 2 - Input settings

[1] This selection is not available on some versions

3.2. Front panel

The front panel includes the following zones, starting from the left:

- Input channels;
- Master section;
- Monitoring section

3.2.1. Input channels

The following picture shows the elements for channels 3 and 4, as an example.

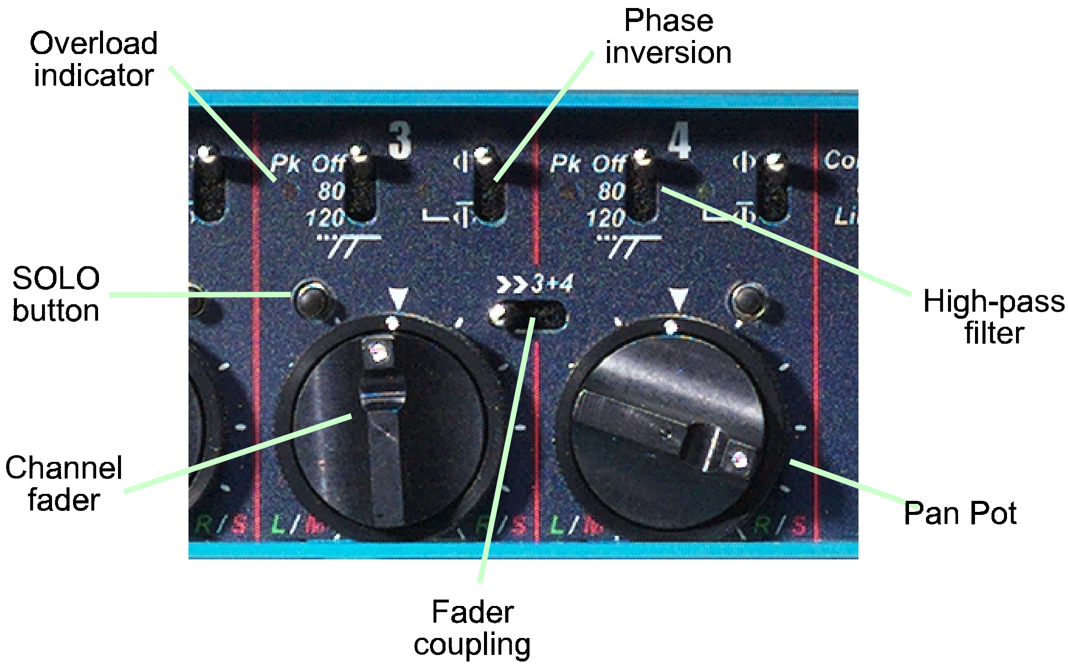


Figure 3 - Input channels (3 and 4)

Notes:

- When the fader coupling switch is set in the rightmost position, channel fader #4 sets the gain for both channels simultaneously, and channel fader #3 is inoperative. Of course, the same holds for the “1+2” fader coupling switch with (respectively) fader #2 and fader #1.
- When pushing the SOLO button on a channel, the corresponding input signal is sent to both monitoring channels, overriding the current monitoring source selection.
- The coloured text on the bottom reminds the panning of the input, depending whether the mixer is operated in normal stereo mode (L/R green text) or in M/S mode (M/S red text).

3.2.2. Master section

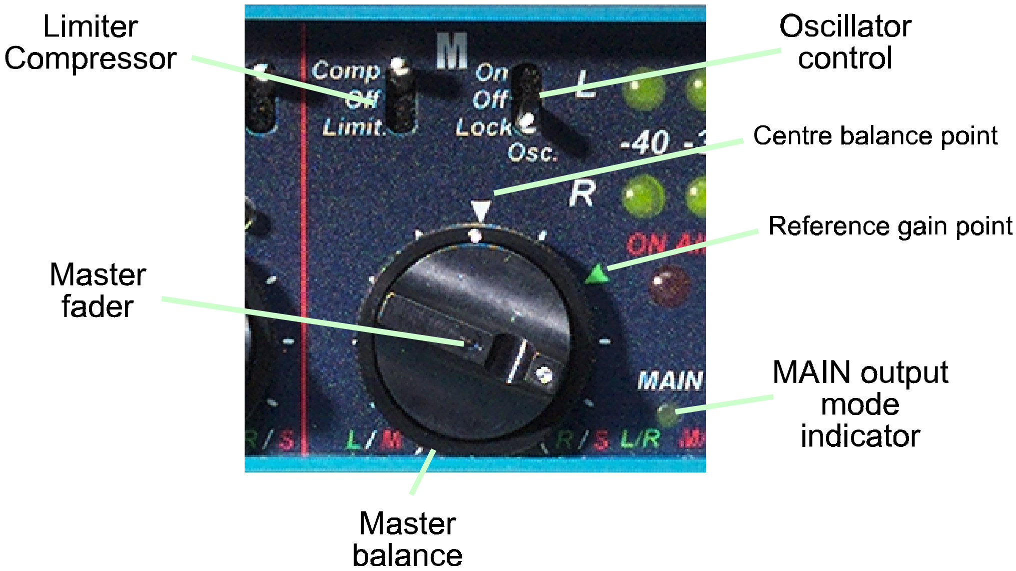


Figure 4 – Master section

Notes:

- The oscillator produces a continuous 1 kHz sine wave on the left channel and an intermittent 400 Hz sine wave on the right channel[1]. The oscillator can be activated either temporarily by sliding up the oscillator switch ("On" position), or permanently by moving down the switch ("Lock" position).
- When the master fader is set at the reference gain point (and the limiter/compressor is off), the MAIN output has a 0 VU level when the internal oscillator is set on.
- Note that the master fader is located **after** the limiter/compressor on the signal chain (see general diagram on page 32).
- The limiter/compressor is a stereo processor, but it is possible (internal configuration) to select independent processing of the two channels.
- The MAIN output mode indicator switches to red when L/R↔M/S matrixing is activated on the MAIN output, while it is green when no matrixing takes place.

[1] By changing the internal configuration, the right channel can be set at 1 kHz, and it can be permanent (instead of intermittent).

3.2.3. Monitoring section

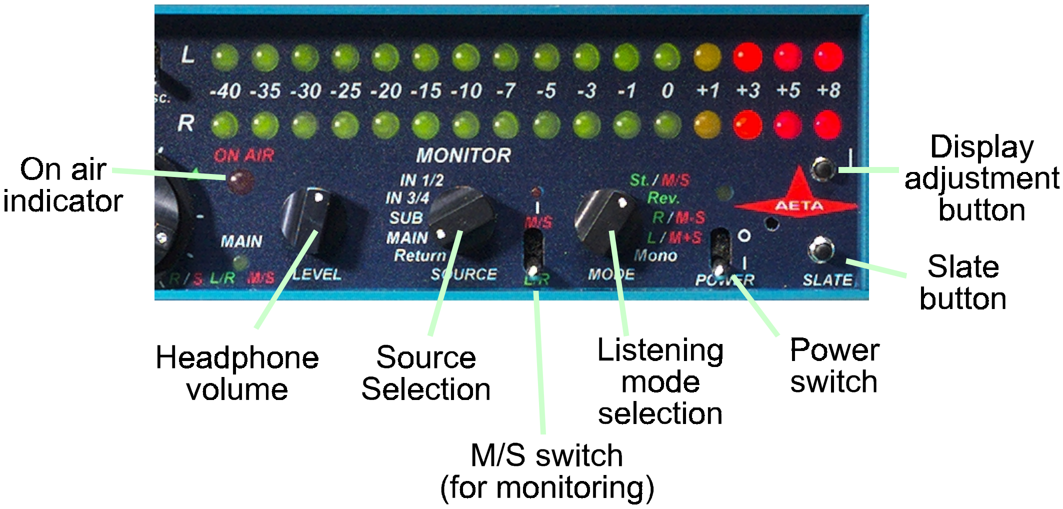


Figure 5 – Monitoring section

Notes:

- The “ON AIR” indicator is activated by the “On-air” control signal from the “Monitor” interface (refer to 2.8, Monitor (MON));
- The M/S switch can be used for optionally matrixing the signal for monitoring, typically for listening in normal stereo when using M/S microphones (see 2.5, Monitoring). A red LED is on when the switch is activated. In such case, the action of the mode selector is indicated by red coloured text near the selector.
- The LED above the power switch lights green when the unit is on. However, the LED flashes red whenever the batteries are low (less than 8.8 V, or 1.1 V per battery), or the external DC power is low (less than 12.8 V[1]).
- The Slate button can be used for activating the talkback microphone. The microphone is located behind a small cavity in the front panel, just besides the “Slate” button.
- The button below the bar-graphs is used for adjusting the display mode and/or the display brightness. The following chapter describes the instructions for using this button.

[1] Note that the threshold is different for the external power

3.3. Adjusting the bar-graph display

As described above in 2.5, the display brightness is adjustable and two level display modes are available:

- Standard bar-graph mode, with peak hold;
- “Dot-only” mode, where only the last peak is displayed.

The display mode and brightness can be set using the push-button near the display.

3.3.1. Display mode selection

When switched on, the mixer is in standard bar-graph display mode. Shortly pressing the button switches the display to “Dot-only” mode, pressing it again brings it back to normal mode.

3.3.2. Display brightness adjustment

The display brightness can be adjusted among five intensity levels, from very low intensity (for use in the dark) to a level high enough to read it in sunlight.

When switched on, the mixer is set with an average brightness. Press and hold down the button in order to scroll the different brightness levels, by decreasing brightness order. Release the button when the desired level is reached.

Note: after reaching the lowest level, the cycle resumes from the highest level.

3.4. Right panel: “Outputs”

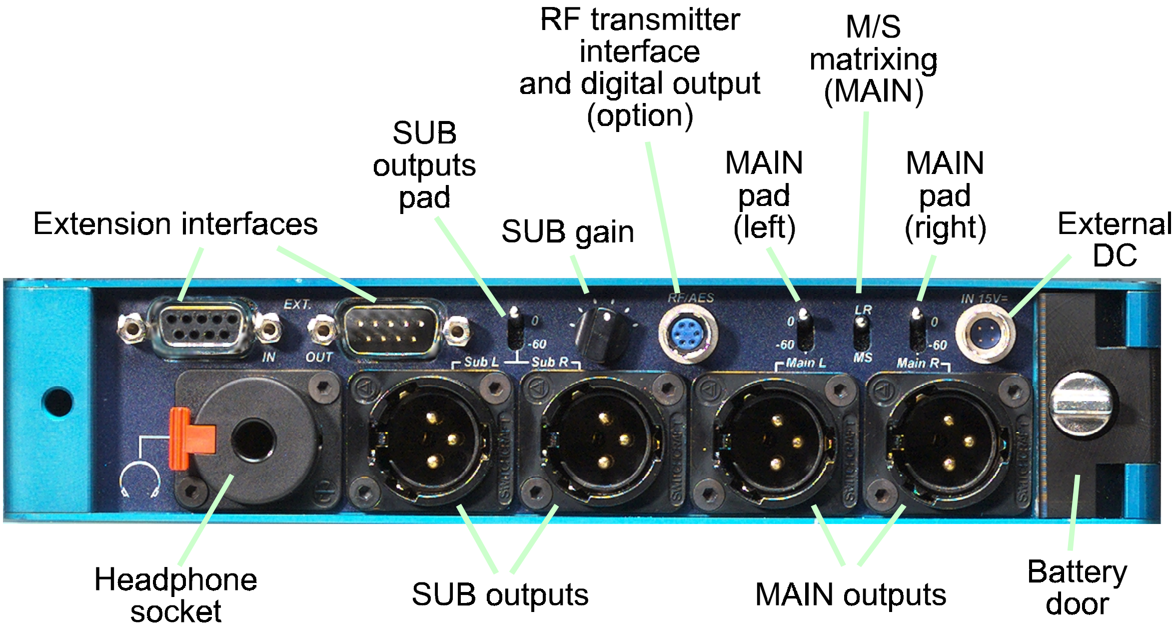


Figure 6 – Right panel

Notes:

- The headphone socket is a locking connector for a ¼” jack; for unplugging the jack, remember to unlock the socket by pressing the button;
- Both SUB and MAIN outputs are available on male XLR sockets. However, only the MAIN outputs are symmetrical, while the SUB outputs are asymmetrical;
- The SUB pad attenuates both left and right outputs at the same time, while separate pads are available for MAIN left and right signals;
- M/S matrixing switch: no matrixing when on the “L/R” position. M/S position can be used for outputting a normal stereo signal while using M/S microphones, or conversely for outputting M/S encoded signals while using the mixer in normal stereo mode. Note that, when monitoring the MAIN output, the signal is picked up after the possible matrixing.
- When the mixer is fitted with the digital output option, this output is available on the RF transmitter link socket. An adapter cord for SPDIF format, included in the option, makes the signal available on a male RCA plug. Alternatively, it is possible to use an adapter cord for AES format with a male XLR plug. Plugging the adapter cord on the mixer activates the digital output module; in order to maximise battery life, it is recommended to disconnect the cord when not using the digital output.
- The external DC socket accepts a DC supply between 13 V and 16 V. The mixer still operates correctly at a voltage down to 4 V, but the “battery low” indicator flashes whenever the external DC voltage is less than 12.8 V.
- The external DC can be plugged in or out at will during operation. Operation is not disturbed provided there are batteries inside the mixer.

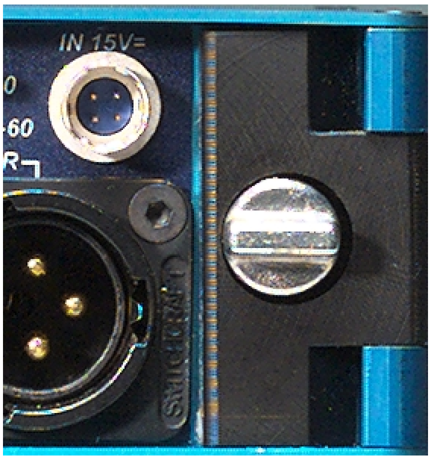
3.5. Replacing the batteries

The mixer operates from 8 LR/AA-size batteries. The following types can be used:

- Disposable batteries; heavy duty alkaline batteries are highly recommended.
- Rechargeable NiCd or NiMH batteries.

To unlock the battery door, simply twist the door lock a quarter of a turn. Then the door unlocks and half opens. To open it and release the batteries, slightly push the door (leave the door lock loose).

When inserting new batteries, the correct orientation of the batteries is shown on the back panel (see the + and – signs inside the battery door as well). To lock the battery door, hold the door closed, and firmly push the door lock while holding it in the proper orientation (that shown on the photograph below), until it locks into place. You may also twist the door lock slightly while pushing it.



For proper operation of the mixer:

- Replace all batteries at the same time for new batteries of identical types; do not mix batteries from different makes or models;
- Check that the batteries are set in their compartment as indicated, with proper orientation;
- Switch off the mixer when the batteries are low and/or the mixer is not in use;
- It is recommended not to leave batteries inside the mixer when it is left unused for a long time.

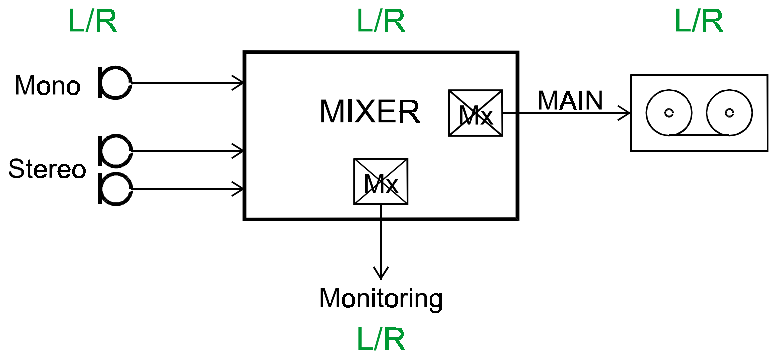
Possible damage induced by the non observance of the above recommendations is not covered by the warranty.

3.6. Using L/R <-> M/S matrixing

This chapter shows examples of situations where L/R↔M/S matrixing can be used and the effect of matrixing in these situations. Further information about the M/S stereo technique can be found in annex 6.1 (Overview of the M/S stereo technique).

3.6.1. Case 1: standard stereo engineering

This is the most common application. In this case, mono microphones or normal (L/R) stereo microphone couples are used, and the output of the mixer is in standard stereo format.

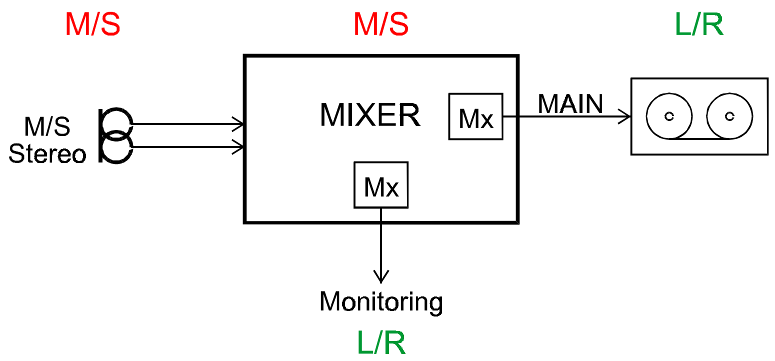


L/R↔M/S matrixing is not needed in such an operating mode.

However, L/R↔M/S matrixing may be used occasionally for monitoring, as a way of visually checking the stereo image consistency on the level meter display. If matrixing the monitored signal, the "left" bar-graph shows the L+R sum, while the "right" bar-graph shows the L-R difference. For common stereo production, the L-R level should be much lower than the L+R level, especially when mono compatibility is desired. On the contrary, L-R level higher than L+R level would warn about a possible problem in sound pick-up (e.g. one microphone in a stereo couple is phase reversed).

3.6.2. Case 2: stereo recording using MS microphones

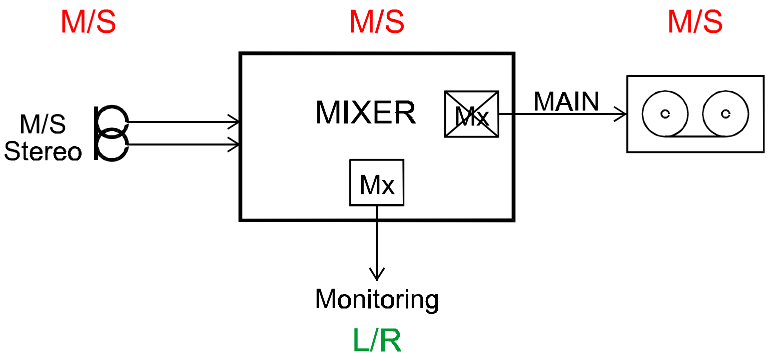
In this case, M/S stereo microphone couples are used, and the output of the mixer is in standard stereo format.



Instead of left and right, the mixer buses actually process M (Mid) and S (Side) signals. In order to output normal stereo signals, L/R↔M/S matrixing must be applied at the MAIN output (switch on the "M/S" position). Matrixing is also needed for monitoring a normal stereo signal on the headphones (front panel switch on the "M/S" position), except for monitoring the MAIN output because it is already decoded to L/R format.

3.6.3. Case 3: MS recording using MS microphones

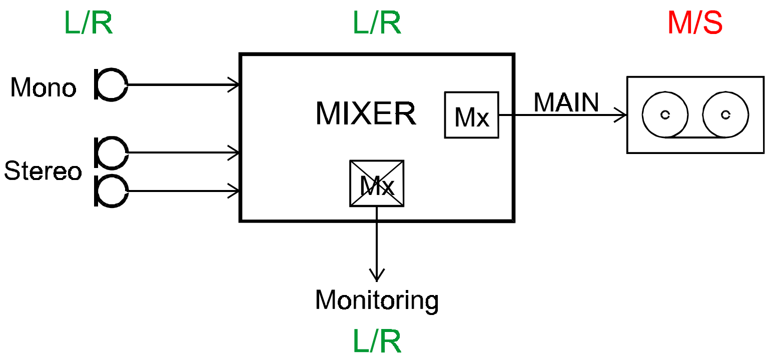
In this case, M/S stereo microphone couples are used, and the output is kept in M/S format for recording. In this way, further stereo image processing is made possible with studio equipment in later production.



The mixer buses process M (Mid) and S (Side) signals. L/R↔M/S matrixing is not applied at the MAIN output (switch kept on the “L/R” position). Matrixing is needed for monitoring a normal stereo signal on the headphones (front panel switch on the “M/S” position).

3.6.4. Case 4: MS recording using classical microphones

In this case, mono microphones or normal (L/R) stereo microphone couples are used, but the output is encoded in M/S format for recording. In this way, further stereo image processing is made possible with studio equipment in later production.



In order to encode the outputs in M/S format, L/R↔M/S matrixing is applied at the MAIN output (switch on the “M/S” position). Matrixing is not needed for monitoring a normal stereo signal on the headphones, except for monitoring the MAIN output, which is encoded in M/S format.

4. First level maintenance – Internal configuration

4.1. Internal description

The following diagram shows the organisation of the mixer, as can be seen from the bottom side if the bottom cover is removed.

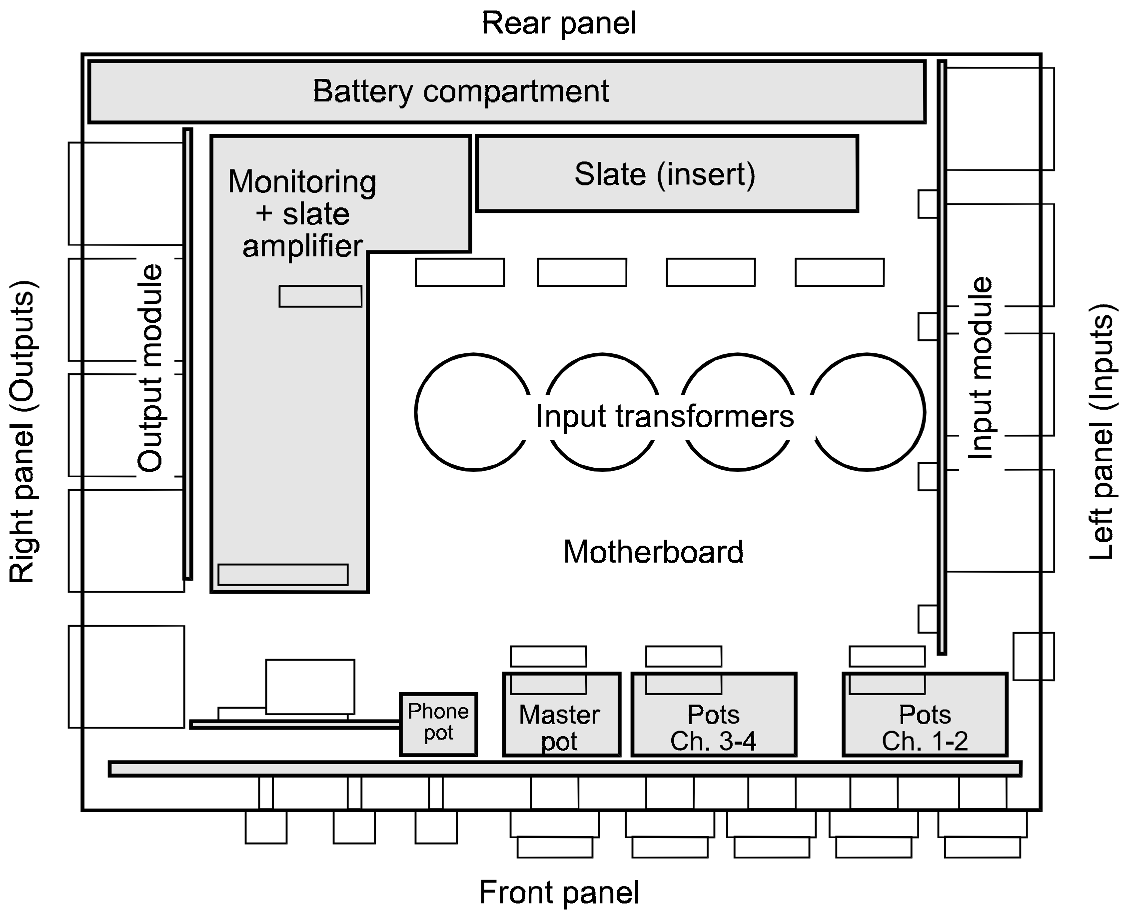


Figure 7 – Internal organisation (view from the bottom)

The main part of the mixer is the motherboard, mounted on the top of the mixer, parallel to the top cover. Various daughterboards and modules are connected to the motherboard:

- Front panel assembly, including all front panel control and display elements;
- Input module on the left side;
- Output module on the right side;
- Monitoring and slate[1] daughterboards mounted directly upon the motherboard.

For easy replacement, the potentiometers are mounted on small daughterboards attached to the front panel module.

[1] When the digital output is installed, the slate daughterboard also includes the analogue to digital converter and the digital output.

4.2. Internal configuration (switches and jumpers)

Most configuration elements are available on the top side of the motherboard. However, a few elements can be accessed from the bottom side.

> **Warning**: whenever opening the unit, only use adequate tools, and take all necessary care to avoid electrostatic discharges (ESD) in the unit. Damage to the internal parts due to non observance of these precautions is not covered by the warranty.

4.2.1. Top side configuration elements

For accessing the elements, remove all screws on the top and put the top cover aside. The motherboard appears as on the following picture. The jumpers and switches are shown with the effect of each position.

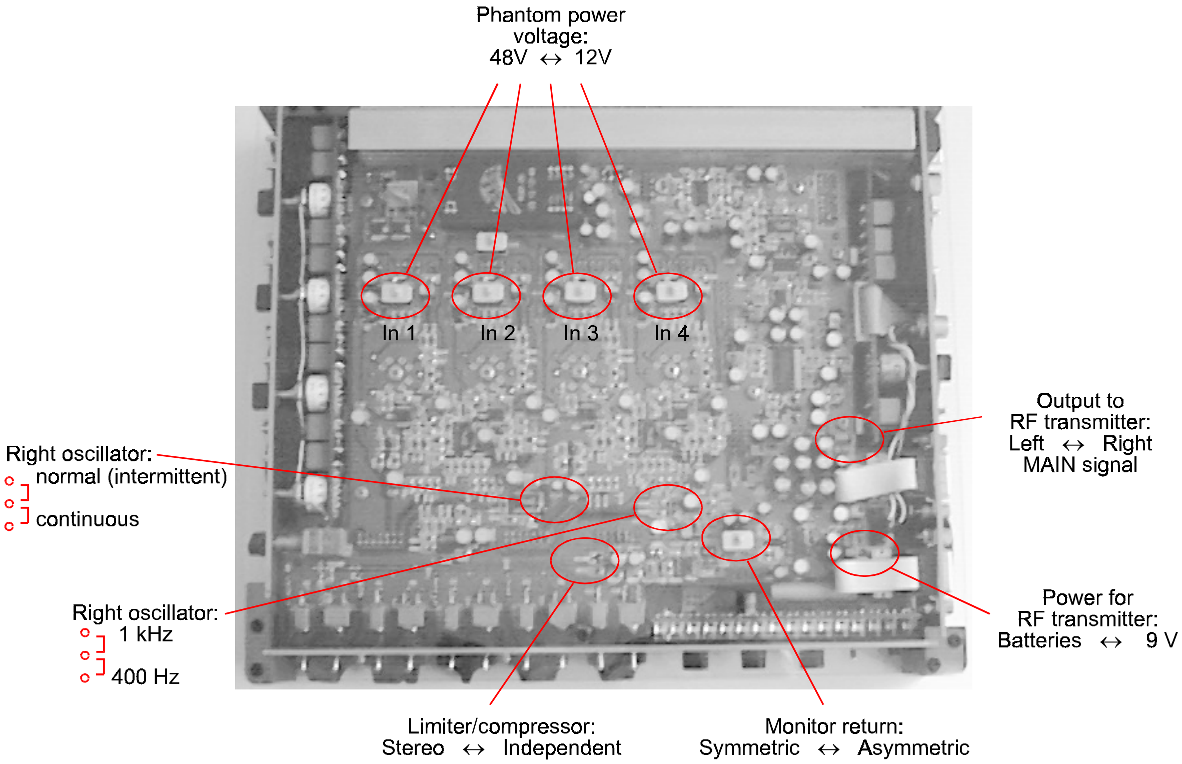


Figure 8 – Internal configuration – top side view

Notes:

- The phantom power voltage can be configured 48 V or 12 V separately for each input.
- Depending on the position of the corresponding jumper, the power output to the RF transmitter is either a regulated +9 V supply or directly the battery power (or the external DC source when available).

When remounting the cover, please check that the flanges are well engaged inside the top cover grooves before tightening the screws.

4.2.2. Bottom side

For accessing the elements, remove all screws on the bottom and put the bottom cover aside[1]. The motherboard appears as on the following picture. The jumpers are shown with the effect of each position.

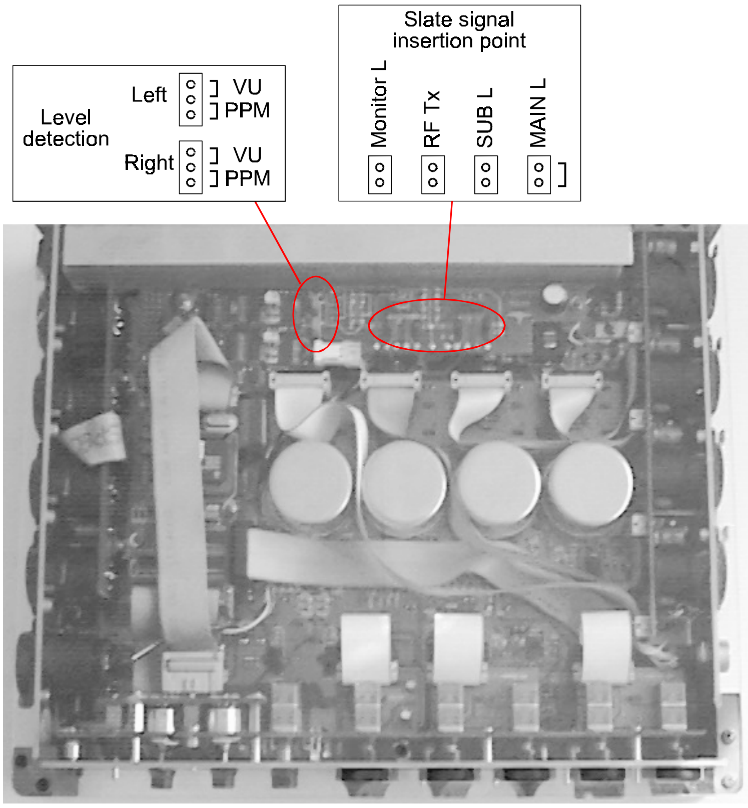


Figure 9 – Internal configuration – bottom side view

Notes:

- Slate signal insertion: the jumper must be set on one among the four allowed positions (or none of them in order to disable the slate function). It is also allowed to select more than one position, as the signal is injected on each output for which the corresponding jumper is set on.
- Please also refer to following chapter 4.2.3 in case the mixer is equipped with a digital output.
- Do not move other jumpers than those described above.

When remounting the cover, please check that the flanges are well engaged inside the cover grooves before tightening the screws.

[1] Caution: only remove one cover at a time.

4.2.3. Configuration of the digital output option

In a MIX2000 featuring a digital output, the “Slate” module (top right on Figure 9 above) also includes the analogue to digital converter and the digital output. The configurable elements on this module appear as shown on the following drawing.

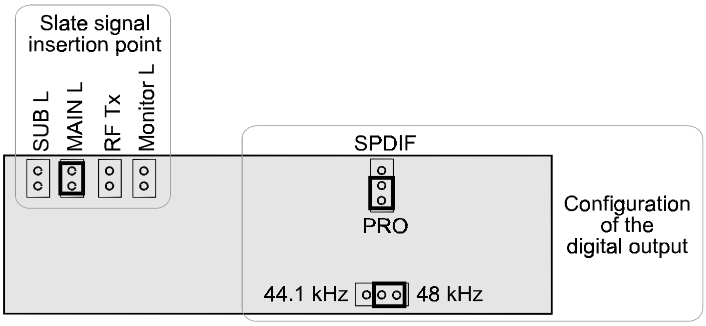


Figure 10 – Configuration of the optional “Slate and digital output” module

(In the example shown, the Slate signal is inserted on the left MAIN output, and the digital output is AES professional format at 48 kHz sampling rate)

The selection of the slate signal insertion point can be found on this module; the jumper must be set on one or more of the available positions (or none if the function is not desired).

The other jumpers select the format of the digital output. The sampling rate can be set at 44.1 kHz or 48 kHz. Besides, the SPDIF/PRO jumper selects the frame format of the digital output:

- “SPDIF”: consumer format, SPDIF or IEC958.
- “PRO”: professional format, compliant with AES3 recommendation;

The proper selection depends on the application. The following table shows the allowed combinations as well as some typical applications. The table also shows the external adapter cable (and hence the output plug) to use, to be consistent with that combination.

Frame format	Sampling rate	Application examples	Adapter cord to be used
SPDIF	44,1 kHz	Recording on a portable DAT or MD recorder	RCA[1]
	48 kHz		
PRO (AES)	44,1 kHz	Recording on a DAT device	XLR
	48 kHz	Recording (professional equipment) Connection to a digital mixing console	

Please note that the 44.1 kHz sampling rate **must** be selected with the SPDIF consumer format. Also note that using a cord other than recommended produces an incorrect format (e.g. SPDIF frame format with AES electrical format on an XLR plug), which leads to unpredictable results on the equipment connected at the output.

[1] This adapter cord is delivered with the digital output option

4.3. Internal adjustments

Warning: all adjustments in the mixer are done on the factory, and should need no later changes. Only qualified personnel should change these adjustments. For access to the adjustable elements, follow the same instructions as for the internal configuration described in the previous chapter.

4.3.1. Top side

Most adjustments are accessible from the top, as visible on the following picture:

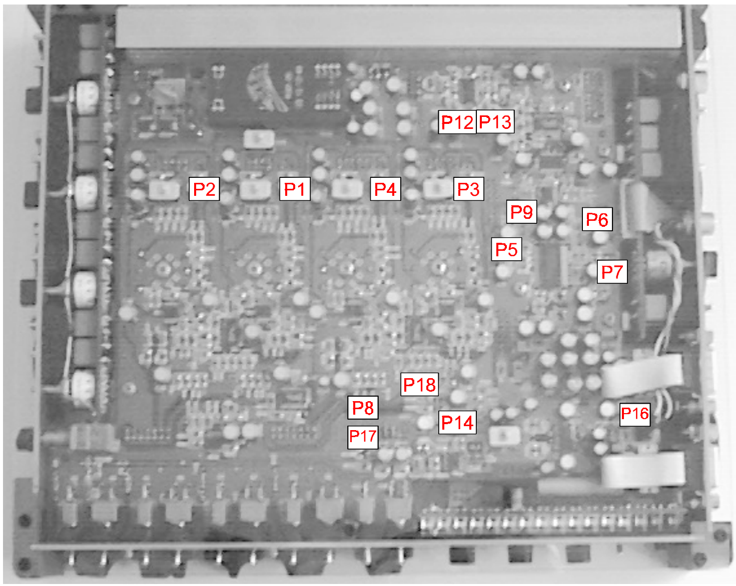


Figure 11 – Internal adjustments – top side view

Function	Reference	Adjustment
Input symmetry adjustment (Common mode rejection)	**P2**	Input symmetry (Channel 1)
	P1	Input symmetry (Channel 2)
	P4	Input symmetry (Channel 3)
	P3	Input symmetry (Channel 4)
Master VCA adjustments	**P6**	Balance centre point
	P8	Compression ratio (left)
	P17	Compression ratio (right)
	P9	Limiter/compressor threshold
MAIN outputs	**P5**	THD (Left)
	P7	THD (Right)
	P13	Output common mode (Left)
	P12	Output common mode (Right)
Oscillator	**P18**	Oscillator level (left)
	P14	Oscillator level (right)
+9 V supply (for RF transmitter)	**P16**	Voltage adjustment

4.3.2. Bottom part

From the bottom side, it is possible to access trimmers on the Monitoring and slate amplifier daughterboard, as visible on the following diagram:

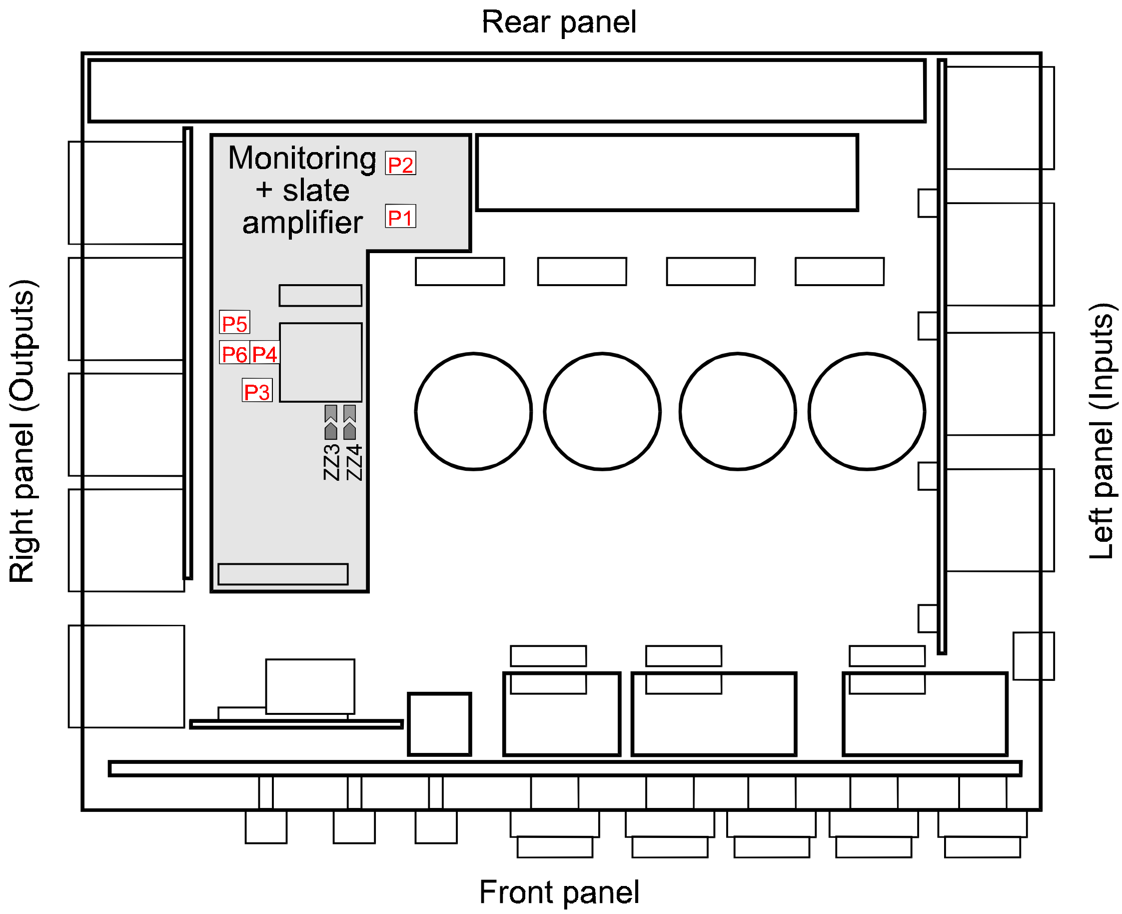


Figure 12 – Internal adjustments – bottom side view

Function	Reference	Adjustment
Programme level meter	**P1**	Gain calibration (Right)
	P2	Gain calibration (Left)
	ZZ3 **ZZ4**	Mode and brightness settings at power on time
Slate microphone amplifier and dynamics processor	**P3**	Overall gain
	P4	Noise gate threshold
	P5	Compressor slope
	P6	Rotation point

By shorting (with solder) ZZ3 and/or ZZ4, it is possible to select for the level meters another start-up mode than the factory setting (■ = solder bridge, - = no solder, open circuit):

Configuration number	Configuration		Display mode on start-up	
	ZZ3	ZZ4	Mode	Brightness
0 (factory)	-	-	Bargraph	Medium
1	■	-	« Dot »	Medium
2	-	■	Bargraph	Maximum
3	■	■	« Dot »	Minimum

5. Technical characteristics

5.1. Interface characteristics

5.1.1. Microphone/line inputs

Format	Symmetrical, transformer isolated
Connectors	XLR female socket (3-pin)
Microphone powering	Phantom supply, 48 V or 12 V[1] Ton-adder[2], 12 V
Transformer isolation	> 500 V_{DC}
Transformer ratio	1 : 6
Input pad attenuation	26 dB
Maximum input level with pad	+31 dBu
Nominal input headroom	> 56 dB
Input gain (sensitivity setting)	+15 to +57 dB by 6 dB steps (8 position rotary switch)
Input impedance	> 2 kΩ
Common mode rejection (CMRR)	≥ 90 dB @ 1 kHz

5.1.2. MAIN outputs

Format	Symmetrical, electronically balanced L/R or M/S encoded
Connectors	XLR male socket (3-pin)
Output pad attenuation	60 dB (separately available on left and right)
Nominal output level	+4 dBu / 0 VU
Maximum output level	+25 dBu
Output impedance	≤ 50 Ω
Output symmetry	≥ 40 dB
M/S matrixing network matching	> 40 dB @ 1 kHz

[1] Depending on internal configuration; refer to chapter 4.2, Internal configuration (switches and jumpers).

[2] Not available on some equipment versions

5.1.3. SUB outputs

Format	Asymmetrical
Connectors	XLR male socket (3-pin)
Output pad attenuation	60 dB (common for left and right)
Nominal output level	+4 dBu / 0 VU
Maximum output level	+20 dBu
Output gain (without pad)	-30 to +0 dB by 6 dB steps (6 position rotary switch)
Output impedance	≤ 50 Ω

5.1.4. Headphone output

Connector	¼" jack socket, locking
Maximum output level	+20 dBu
Nominal load impedance	≥ 16 Ω

5.1.5. External DC

Connector	4-pin HiRose socket (HR10-7R-4P) See pin-out in following table
Nominal voltage	≥ 13 V
Minimum voltage for operation	5 V
"Low battery" threshold	12.8 V
Maximum permitted voltage	16 V

The connector has following pin-out:

<table>
<tr><th>Pin</th><th>Function</th></tr>
<tr><td>1</td><td rowspan="2">- External DC</td></tr>
<tr><td>2</td></tr>
<tr><td>3</td><td rowspan="2">+ External DC</td></tr>
<tr><td>4</td></tr>
</table>

5.1.6. RF micro-transmitter interface

Connector	6-pin HiRose socket (HR10-7R-6S) See pinout in following table
Audio output format	Asymmetrical, line level
Output level	-2 dBu nominal
Source impedance	1 kΩ
DC power output	+9 V or battery voltage (Max. 100 mA on +9 V)

The connector has following pin-out:

Pin	Function
1	Audio output
2	NC
3	NC
4	DC power supply for the RF transmitter
5	Power and signal ground
6	NC

The audio output is the MAIN left or right signal depending on the internal configuration; for details, refer to chapter 4.2, Internal configuration (switches and jumpers).

Pins 2, 3 and 6 should remain unconnected; they are used for the digital output when this option is installed in the MIX2000 (see 5.1.9, digital output).

5.1.7. Monitor interface

Connector	12-pin HiRose socket[1] (HR10-10R-12S) See pinout in following table
Audio input/output format	Line level
MON Return gain	Adjustable 0 to -∞ dB (fader near the Monitor socket)

The connector has following pin-out:

Pin	Function	Direction
1	ON AIR relay command (to the mixer); +12 V	IN
2	Right output (MAIN R signal)	OUT
3	Left output (MAIN L signal)	OUT
4	Mono output (MAIN L+R)	OUT
5	Signal ground	-
6	+ MON Return Left	IN
7	- MON Return Right	IN
8	- MON Return Left	IN
9	Signal ground	-
10	+ MON Return Right	IN
11	+ SLATE	OUT
12	- SLATE	OUT

5.1.8. Extension inputs/outputs

EXT IN Connector	9-pin female sub-D socket See pinout in following table
EXT OUT Connector	9-pin male sub-D socket See pinout in following table

[1] As an option, a 10-pin connector (RM15TRD-10S) can be installed. Consult AETA AUDIO for such option.

The EXT IN connector has following pin-out (NC = no connect):

Pin	Function
1	Input to right mixer bus
2	Signal ground
3	Input to left mixer bus
4	NC
5	Connected to DC power (batteries)
6	NC
7	NC
8	NC
9	NC

The EXT OUT connector has following pin-out (NC = no connect):

Pin	Function
1	MAIN R signal
2	Signal ground
3	MAIN L signal
4	Signal ground
5	Connected to DC power (batteries)
6	Channel 1 signal (pre-fader)
7	Channel 3 signal (pre-fader)
8	Channel 2 signal (pre-fader)
9	Channel 4 signal (pre-fader)

5.1.9. Digital output (option)

When the option is installed, the digital output is available on the same socket as the RF transmitter interface. This socket then has following pin-out:

Pin	Function
1	Audio output (to RF transmitter)
2	**Digital output (+)**
3	**Digital output (-)**
4	DC power supply for the RF transmitter
5	Power and signal ground
6	**Digital module power control signal**

The digital output module is only powered when pin 6 is shorted to ground (pin 5). This link is included in the adapter cord used for the digital output, so this module is not powered when the cord is disconnected, saving power when the digital output is not used.

The digital output available on pins 2 and 3 is electrically AES format. However, the digital output adapter cord delivers the signal on a standard plug, and converts it to the appropriate electrical format:

With the SPDIF adapter cord (included with the digital output option), the output features following characteristics:

Standard	IEC958 (a.k.a. SPDIF, consumer format)
Format	Asymmetric, and transformer isolated
Connector	RCA male plug
Output level (on 75 Ω load)	1 V c-c
Output impedance	75 Ω

With the AES adapter cord (available as a separate accessory), the output features following characteristics:

Standard	AES3 (professional)
Format	Symmetric, transformer isolated
Connector	XLR male plug (3-pin)
Output level (on 110 Ω load)	5 V c-c
Output impedance	110 Ω

The cord used must be consistent with the mixer internal configuration (see 4.2.3, Configuration of the digital output option).

In any case, the maximum digital level (0 dBFS) is reached when the MAIN analogue output has a +10 VU level (+14 dBu). In other words, **the 0 VU (+4 dBu) reference level on the analogue output corresponds to –10 dBFS on the digital output**.

For proper monitoring, it is highly recommended to set the level meter for PPM ballistics when using the digital output.

5.2. Internal characteristics

5.2.1. Internal oscillator

The internal oscillator produces a steady 1 kHz sine wave on the left channel and a 400 Hz intermittent sine wave on the right channel.

Optionally, by changing the internal configuration, the right channel can be selected to be the same 1 kHz frequency as the left channel. It can also be set for steady operation, like the left channel.

5.2.2. Compressor/limiter

The master channel includes a selectable limiter/compressor, located before the master fader and balance. The limiter/compressor can run in stereo mode or independently on the two channels, depending on the internal settings.

In both limiter and compressor modes, the level threshold (or knee) is +9 dBu (+5 VU).

The compressor rate is 2:1 (factory setting; adjustable internally, see 4.3, Internal adjustments), while the limiter slope is more than 10:1.

5.2.3. Level detection

The level metering function can operate with VU or PPM ballistics, depending on the internal configuration (see 4.2, Internal configuration (switches and jumpers)).

The reference level for level metering (0 on the meter scale) is 0 VU (+4 dBu).

5.2.4. Gains and levels

The diagram enclosed in annex 6.3 (in the end of this manual) indicates the gain of each stage in the signal chain, as well as the nominal and maximum levels in each location.

5.3. System performance

Total maximum gain (Input to MAIN output)	+96 dB
Noise referred to input	< -126 dBu
Amplitude – frequency response	20 Hz – 20 000 Hz ± 0.3 dB
THD + Noise	< -80 dB (0.01 %) @ 1 kHz
Crosstalk between channels	< -80 dB[1]

[1] Except between members of a channel pair (as 1 and 2, or 3 and 4); in such case, pan pot leakage (< -50 dB) dominates in the measurement

5.4. Power supply

The unit operates either from internal batteries or an external DC power supply.

When using internal batteries, the mixer operates from 8 LR6/AA-size batteries. The following types can be used:

- Disposable batteries; heavy duty alkaline batteries are highly recommended.
- Rechargeable NiCd or NiMH batteries.

As external power, either an AC/DC adapter with adequate output voltage, or an external battery pack (e.g. NP1 type packs) can be used.

In the particular case of an external battery pack, it is worth installing batteries in the unit also, as they can provide backup in case the external battery gets discharged. At this moment, the operator may disconnect the discharged external battery pack, replace the pack and reconnect it to the mixer. During this replacement, the mixer will still operate, without any noise, because it switches over automatically to the internal batteries while the external source is interrupted.

The power consumption and the autonomy of the mixer highly depend on the operational conditions, like the type and number of phantom powered microphones, etc. However, the following guidelines can help predict battery life:

- When using only dynamic microphones (no phantom powering), more than 12 hours of operation can be expected from a new set of alkaline batteries.
- In the same conditions, the mixer draws less than 200 mA from an external 14.4 V DC supply (like e.g. a 12-element NiCd battery pack).
- The unit consumes nearly constant <u>power</u> (not constant current) when the input DC voltage decreases; this means that the current drain increases while the batteries get low.

5.5. Dimensions and weight

Dimensions	240 x 190 x 43 mm (9.5” x 7.5” x 1.7”)
Weight	< 2 kg (without batteries)

5.6. Environment

The mixer operates from –20 °C to +50 °C ambient temperature (-4 °F to 122 °F).

It complies with the EC directives for safety and EMC.

- Safety: compliance with EN60950
- Susceptibility: compliance with EN50082-1
- EMI: compliance with EN55022 (class B)

5.7. Versions – Options

The mixer is available with some equipment options:

- A version without T12 microphone powering capability is available on request; in this version, the microphone power switch only has two positions (Dynamic or phantom power).
- Also available on request, the unit can be delivered with a 10-pin, HiRose RM15TRD-10S connector for the Monitor interface.

Besides, the unit can be provided with a digital audio output. With this option, the MAIN output is converted to digital (24 bit resolution) and available in SPDIF format thanks to an adapter cable which is delivered with the option. An adapter cable with AES/EBU output is also available as an accessory (see further).

This option is especially interesting when using the unit with a portable DAT recorder, as the mixer can be seen as a high performance front-end for the DAT recorder, including all necessary high quality audio processing from the microphones to a digital signal for recording with the DAT recorder.

5.8. Accessories

Various accessories are available, including:

- PORTABRACE carrying bag, available in different versions;
- Specific adapter cables, for easy connection to e.g. camcorders, portable RF transmitters, etc.
- Adapter cable for AES/EBU output, for use with the digital output option.
- Tool kits and spare module sets, etc.

Please consult AETA AUDIO or our dealers for detailed information on these accessories.

6. Annexes

6.1. Overview of the M/S stereo technique

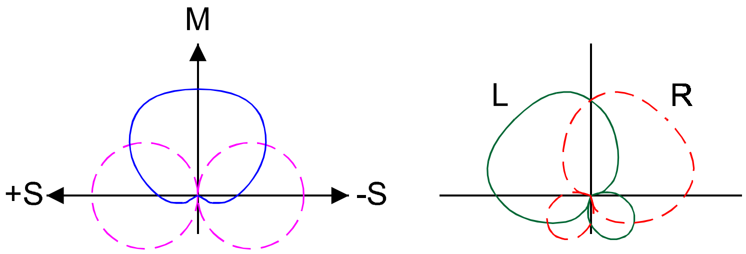
6.1.1. Introduction

The M/S stereo technique provides the modern sound producer an articulate yet versatile stereo image while at the same time maintaining an uncompromised discrete mono-compatible signal. The M/S system is based on combining the signals from a forward-facing microphone. (the Mid or Mono signal) with a laterally oriented figure-8 microphone (the Side or Stereo signal) via a special matrix. Additional flexibility can be had if the Mid microphone has a remote pattern control.

6.1.2. Principles

The M microphone "hears" the principal portion of the sound field and provides the central monophonic pickup. The S microphone responds primarily to directional information and ambience because it is positioned to be most sensitive to the sides and least sensitive to the central portion of the sound field.

These M and S signals are combined in a *sum-and-difference* matrix **(M+S** and **M-S)** to produce the individual left and right signals for conventional stereo. By changing the relative contributions from these microphones, a variety of sound perspectives from monophonic to a very out of phase *hyper-stereo* can be created.



M+S=L and **M-S=R**

In summed mono [L+R] the S component disappears:

(M+S) + (M-S) = 2M

Since the matrix can be introduced at any point in the recording or post-production process, the M/S technique affords complete control of ambience and stereo perspective even during the last stages of the final mix.

6.1.3. Advantages of the M/S technique

Predictable Mono/Stereo compatibility

Mono/Stereo compatibility is of great importance with today's multiple-format releases. Only M/S stereo can serve the upscale stereo audience without compromising the sound for the majority who still listen in mono.

Automatic ambience reduction in mono

Good intelligibility requires less ambience in a monophonic mix, while pleasing stereo requires more ambience and reverb. The M/S technique resolves this dichotomy: when the stereo signal is summed to mono, the S component cancels, reducing the ambience. this way the M/S technique naturally provides more ambience in stereo and less in mono. This reduction of ambience maintains high intelligibility for the monophonic listeners.

6.1.4. Applications for motion picture and broadcast sound production

Broadcasting: stereo compatibility for ENG, documentaries and sports

M/S stereo employs an easy to use, single-point stereo microphone technique that's ideal for capturing "atmosphere", crowd reactions and sound effects. It's "liveness" provides a real sense of "being there".

Entertainment, drama and commercials

By recording dialog, background, SFX and audience reaction as discrete M and S signals on separate tracks, the M channel can be used as conventional mono. The S channel can then be used separately as an ambient perspective. Later it can be matrixed with the M component for creating project-perfect stereo.

Music: Single-Point Placement

M/S stereo captures the natural sonic perspective of any "acoustical" ensemble: classical, folk, ethnic, jazz or even "unplugged" rock. The M/S technique is also convenient for tracking choirs, background groups, drum kits or even logging room tone.

Motion picture sound

Because most movies will eventually be broadcast and/or be released as videos, all of the mono compatibility concerns of broadcast audio apply to motion picture sound production.

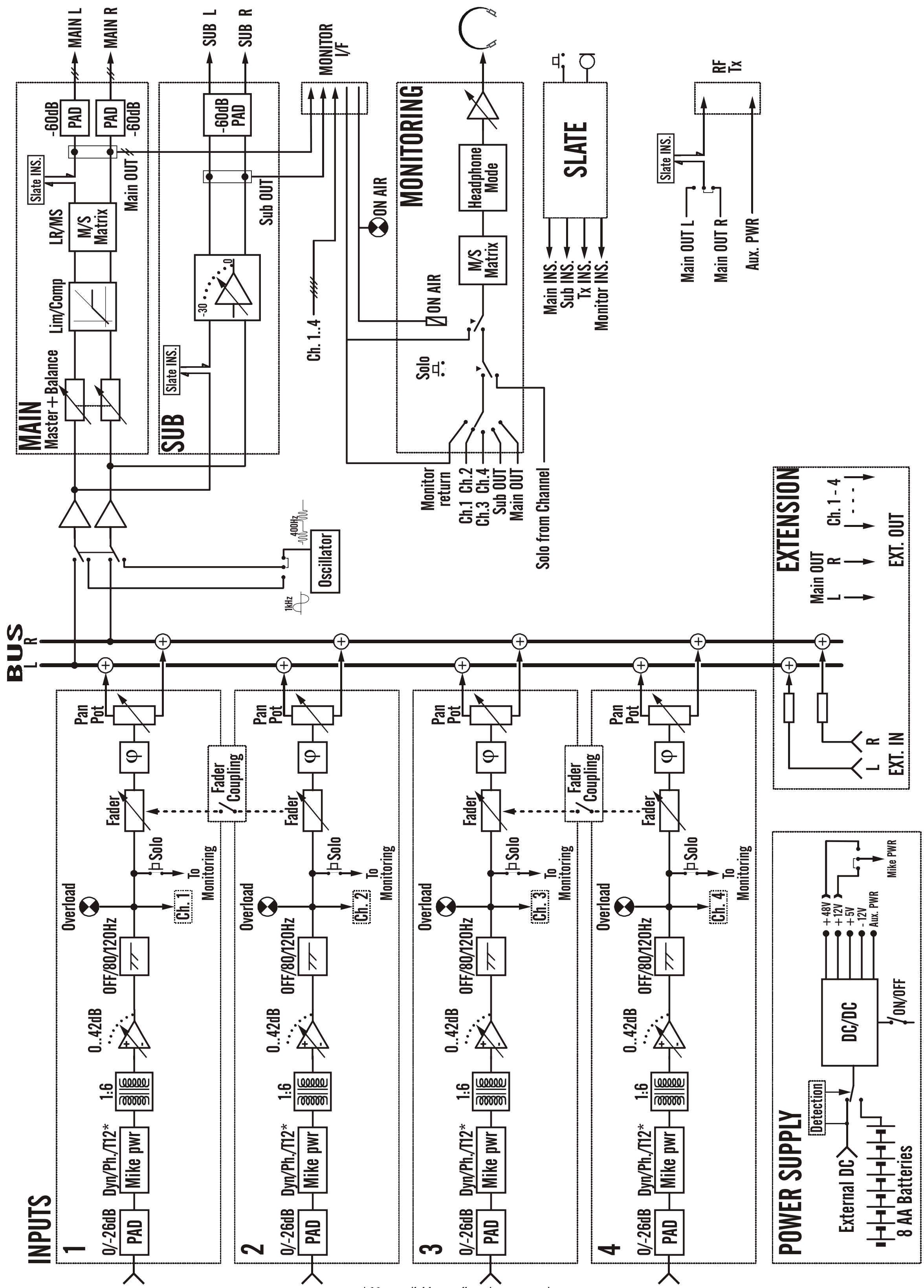
Sound Effects:

The M/S technique is used by leading sound effects houses and video facilities. Another approach is to matrix from M/S to L/R during the original recording, then in post production use matrix to reconstruct the original stereo perspective and adjust it to fit the mix.

6.1.5. Summary

M/S stereo allows proven predictable microphone techniques to be used for the stereo pickup - quickly and reliably. Stereo image width can be adjusted at any time to fit the picture perspective, while always preserving the original discrete mono sound. All this flexibility derives from a simple single-point microphone setup with single-knob control of perspective.

6.2. General synoptic diagram



* Not available on all equipment versions

Figure 13 – General diagram of the mixer

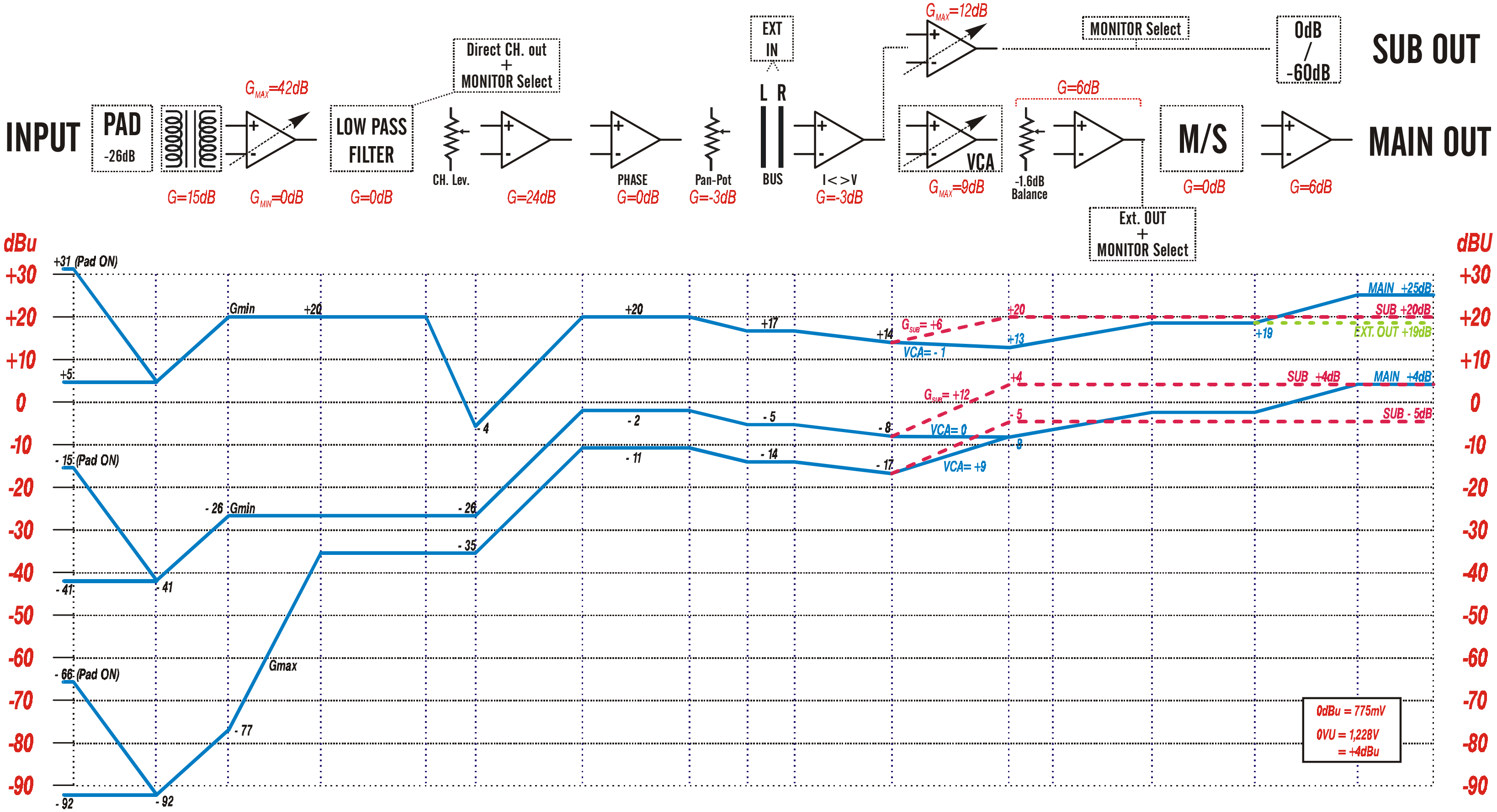


Figure 14 – Gain and level diagram